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United States Patent [19]

Friedman

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[54] **COMMUNICATION SYSTEM**

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[51] **Int. Cl.**[7] H04B 10/18; H04L 27/01
[52] **U.S. Cl.** **375/232**
[58] **Field of Search** 375/232, 233,
375/234, 235, 236, 229; 708/300, 322, 323

[56] **References Cited**

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[57] **ABSTRACT**

A method and system for providing adaptive filtering in a communication system. The method and system modify coefficients of a finite impulse response filter fed by a sequence of digital samples in accordance with an error signal in floating point format. The floating point error signal includes only a sign bit and an exponent term. The exponent term is added to an exponent term of an adaptation coefficient to produce a composite error signal. The adaptive filter is used as a linear adaptive equalizer and as an echo canceler.

21 Claims, 12 Drawing Sheets
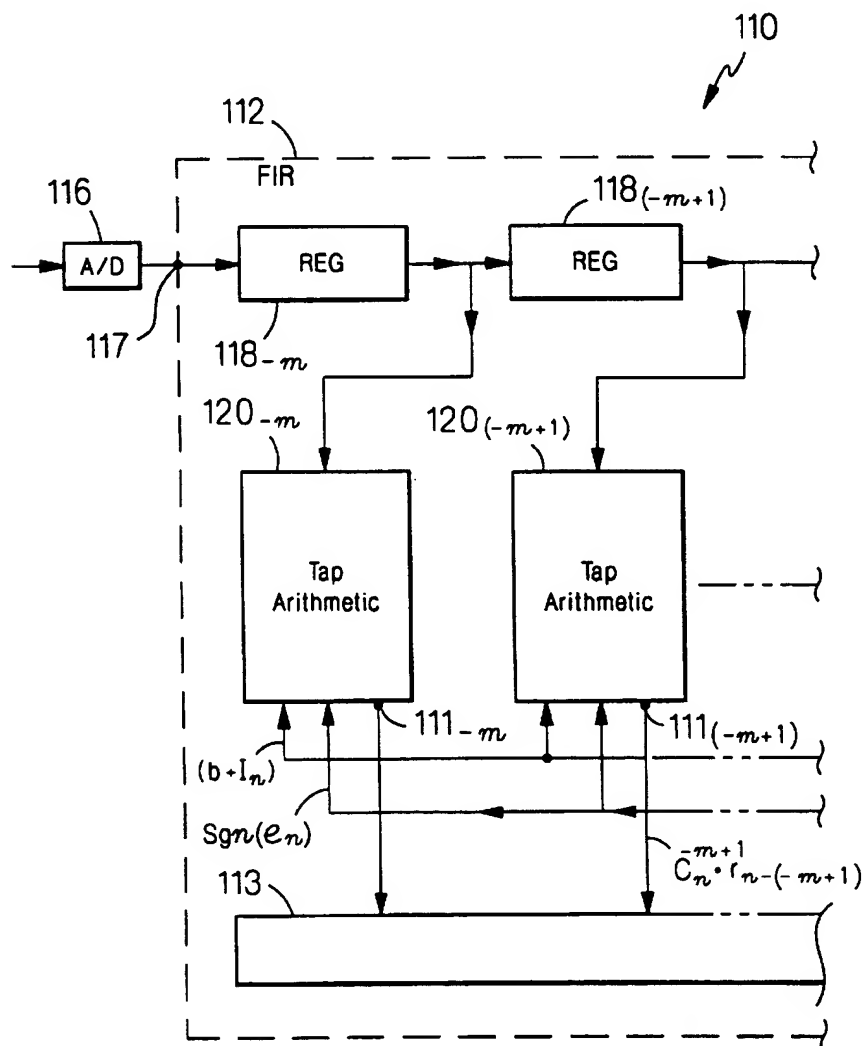
trained
FZR coffecient
A § φ adj
to match
channel

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DOCUMENT-IDENTIFIER: US 6118814 A

TITLE: Communication system

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Number of Claims - NCL (1):
 21

Brief Summary Text - BSTX (3):
   As is known in the art, adaptive filters have been used in communication
systems to provide such functions as channel **phase and amplitude dispersion
equalization** and channel echo cancellation. For example, in transmitting
information to a receiver, such transmitted information passes through a
communication channel. The communication channel may be a pair of wires,
(e.g., a so-called twisted pair such as used in telephones, a fiber optic
cable, a coaxial cable) or may be a designated part of the electromagnetic
spectrum, such as a radio or wireless communication channel. Further, as is
also known in the art, prior to passing the information into the channel the
information undergoes some form of modulation. One form of modulation is pulse
amplitude modulation (PAM). With pulse amplitude modulation (PAM) samples of
the information are taken and converted into pulses having amplitudes
corresponding to the amplitudes of the sampled information signal. One form of
PAM is baseband PAM. Baseband PAM is commonly used for wire pair communication
channels where the signal spectrum is allowed to extend down to zero frequency
(d.c.). Another form of PAM is passband PAM, commonly used in fiber, coaxial,
and wireless communication channels.

Brief Summary Text - BSTX (15):
   Thus, it follows that if the symbol sequence transmitted and filtered by FIR
12 (i.e., the output of FIR 12) and the symbol sequence, an, produced by the
slicer 14 are the same, the output of the slicer 14, a.sub.n, properly
represents the transmitted data and the error signal, e.sub.n, is zero. If, on
the other hand, the error signal, e.sub.n, is not zero, the FIR 10 adjusts the
coefficients c.sup.-M.sub.n through c.sup.+M.sub.n, in a feedback loop nulling
arrangement to drive the error signal, e.sub.n, towards zero. Thus, ideally at
the end of the training mode, the registers 18.sub.-M through 18.sub.+M will
store coefficients c.sup.-M.sub.n through c.sup.-M.sub.n, respectively, to
produce an error signal, e.sub.n, of zero and therefore the characteristics of
the **FIR 12 will be "matched"** to the characteristics of the transmission
channel. Thus, once the a priori agreed upon training mode has elapsed, the
signal on line 25 is changed and the slicer 14 output is gated out of the
equalizer 10 via gate 38 and the slicer 14 output is used by the receiver
during the normal operating mode. If any error signal, e.sub.n, results from
changes in the characteristic in the channel, the error signal, e.sub.n, after
modification by the adaptation coefficient, .beta., (i.e. .beta.e.sub.n) is
used as a feedback signal to adjust the coefficients c.sup.-M.sub.n through
c.sup.+M.sub.n so that the error signal e.sub.n is driven towards zero, i.e, so
that the equalizer 10 removes the effects of channel noise and intersymbol
interference producing effects.

Detailed Description Text - DETX (12):

Thus, it follows that if the symbol sequence transmitted and filtered by FIR 112 (i.e., the output of FIR 112) and the symbol sequence produced by the slicer 114, a_n, are the same, the output of the slicer 114 properly represents the transmitted data and the error signal, e_n, is zero. If, on the other hand, the error signal, e_n, is not zero, the FIR 110 adjusts the coefficients c_{-M_n} through c_{+M_n}, respectively, in a feedback loop nulling arrangement to drive the error signal, e_n, towards zero. Thus, ideally at the end of the training mode, the registers 118_{-M} through 118_{+M} will store coefficients c_{-M_n} through c_{+M_n}, respectively, to produce an error signal, e_n, of zero and therefore the characteristics of the **FIR 112 will be "matched"** to the characteristics of the transmission channel 103. Thus, once the a priori agreed upon training mode is completed, the signal on line 125 changes and the slicer 114 output is gated out of the equalizer 110 via gate 138 and the slicer 114 output is used by the receiver during the normal operating mode. Further, the error signal, e_n, will be the difference between the slicer 114 output and the slicer 114 input (i.e., the output 115 of the FIR 112). If any error signal, e_n, results from changes in the characteristic in the channel 103, the error signal, after modification by the adaptation coefficient, β, (i.e. βe_n) is used as a feedback signal to adjust the coefficients c_{-M_n} through c_{+M_n}, respectively, so that the error signal e_n is driven towards zero, i.e, so that the equalizer 110 removes the effects of channel 103 noise and intersymbol interference.

Claims Text - CLTX (20):

9. A method for **equalizing amplitude and phase** dispersion in a communication channel, such method comprising the step of modifying coefficients of a finite impulse response filter fed by a sequence of digital samples of information received from the channel in accordance with an error signal provided to the filter only by the sign bit and exponent term of the floating point representation of such error signal.

Claims Text - CLTX (55):

21. The system recited in claim 20 wherein the finite impulse response filter comprises a plurality of taps, each tap comprising: